

Cubase SX/SL Effect Parameters



Cubase • SX/SL



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DoubleDelay



This effect provides two separate delays that can be either tempo based or use freely specified delay time settings. Cubase SX/SL automatically provides the plug-in with the tempo currently used in the project.

The parameters are as follows:

Parameter	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect. If DoubleDelay is used as a send effect, this should be set to maximum (100%) as you can control the dry/effect balance with the send.
Tempo sync on/off		The buttons above the two Delay Time knobs are used to turn tempo sync on or off for the respective delay. If set to off (the buttons are white) the delay time can be set freely with the Delay Time knobs, without sync to tempo.
Delay Time 1	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0-9999ms	This is where you specify the base note value for the delay if tempo sync is on. If tempo sync is off, it sets the delay time in milliseconds.
Delay Time 2	As above	As above.
Feedback	0-100%	This sets the number of repeats for both delays.
Tempo Sync 1	x1 to x10	The note value multiplier for the first delay unit.
Tempo Sync 2	x1 to x10	As above, but for the second delay unit.
Pan1	-100 to 100%	This sets the stereo position for the first delay.
Pan2	-100 to 100%	This sets the stereo position for the second delay.

You can also change parameters in the graphic display window. This works as follows:

- If tempo sync is on, you can set the Tempo Sync 1 parameter by dragging the light blue handle left and right.
When tempo sync is off, this sets the Delay Time 1 parameter.
- You can set the Pan 1 parameter by dragging the light blue handle up and down.
- The dark blue handle works in the same way but for the corresponding second delay parameters.

ModDelay



This is a delay effect that can either be tempo-based or use freely specified delay time settings. The delay repeats can also be modulated. The parameters are as follows:

Parameter	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect. If ModDelay is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Tempo sync on/off		The button above the Delay Time knob is used to turn tempo sync on or off. If set to off (white button) the delay time can be set freely with the Delay Time knob, without sync to tempo.
Feedback	0-100%	This sets the number of repeats for the delay.
Delay Time	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0-9999ms	This is where you specify the base note value for the delay if tempo sync is on. If tempo sync is off, it sets the delay time in milliseconds.
Tempo Sync knob	x1 to x10	This is the note value multiplier for the delay when tempo sync is used.
DelayMod.	0-100%	This controls the pitch modulation rate for the delay effect.

DaTube



This effect emulates the characteristic warm, lush sound of a tube amplifier. It is usable both as an insert effect and a send effect:

Parameter	Values	Description
Drive	0 - 100%	Regulates the pre-gain of the “amplifier”. Use high values if you want an overdriven sound just on the verge of distortion.
Balance	0 - 100%	This controls the balance between the signal processed by the Drive parameter and the dry input signal. For maximum drive effect, set this to its highest value.
Output	-∞ - 0.000	Adjusts the post-gain, or output level, of the “amplifier”.

Overdrive



Overdrive is a distortion-type effect, emulating the sound of a guitar amplifier. A selection of factory styles is available. Note that these are not stored parameter settings, but different basic overdrive algorithms, with the style names indicating the basic character of each algorithm. The parameters are as follows:

Parameter Values		Description
Input	-15 to +15dB	Sets the input level.
Output	-15 to +15dB	Sets the output level. As overdrive generates harmonics, it increases the level of the processed signal. You can use the Output fader to compensate for the level increase.
Speaker simulation	On/Off	Simulates the sound of a speaker cabinet.
Factory Styles	Warm, Chordy, Magic OD, Fat Drive, Woody, Bluesy	Select one of six presets, which can be used as they are or as a basis for further “tweaking”.
Bass	-15 to +15dB	Tone control for the low frequencies.
Mid	-15 to +15dB	Tone control for the mid frequencies.
Hi	-15 to +15dB	Tone control for the high frequencies.
Drive	0-100%	Governs the amount of overdrive. You can also adjust this by clicking and dragging in the display.

QuadraFuzz



QuadraFuzz is a high quality distortion effect allowing control over the level divided into four frequency bands both before and after distortion. This high level of control can create a very wide selection of distortion effects, ranging from subtle to extreme. The user interface consists of two windows.

- The main window features four Filterbank controls, the master Gain and Output controls and a preset selector.
- In the editor window (which is opened by clicking the “Edit” button in the lower right corner) the main feature is a frequency band display. This is where you set the width of the frequency bands as well as their level before distortion.

How does QuadraFuzz work?

Here's a short description of the three major factors that determine how QuadraFuzz sounds, and where you find the corresponding controls:

- The signal volume control *before* distortion.
You can use the Gain control on the left side of the QuadraFuzz main window to control the overall input level of the signal that is fed into the distortion stage. The signal is split up into four frequency bands in the editor window, with adjustable width and level controls. These control the input level before distortion.



- The distortion type, based on a selectable distortion characteristic.

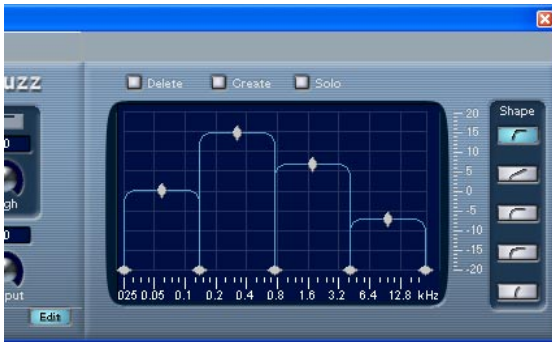


- The signal volume control *after* distortion.
The Output control on the right side of the QuadraFuzz main window controls the overall output level. In addition, the Filterbank controls in the same window allow you to raise or lower the output volume of each separate frequency band that was defined in the editor window.



Editing in the frequency band display

The signal is divided into four frequency bands before being passed to the distortion stage, as explained earlier. You adjust the level and width of these bands in the frequency display.



The frequency band display

Two value scales as well as a number of rhomb- and diamond-shaped handles are available.

- The diamond-shaped handles at the bottom are used to define the corner frequencies of the different frequency bands.
- By using the rhomb-shaped handles on top of each frequency band you determine its relative level before distortion.
- The horizontal value scale below the Frequency band display indicates frequency. The maximum value on this scale corresponds to half the sample rate of the audio file used (Nyquist theorem).
- The vertical value scale to the right shows the approximate level of an edited frequency band.
- If you click and hold on one of the handles, its current value is displayed. Depending on the handle type, corner frequency or level is shown.
- The corner frequency handles can be moved by dragging horizontally. The level handles can be moved by dragging them up or down.
- To reset a level handle to 0 dB, hold down the [Shift] key on your computer keyboard and click on the handle.
- If you hold down the [Ctrl]/[Command] key and move a handle, the values will change in smaller steps.
- The “Solo” button above the frequency band display allows you to monitor individual frequency bands.

If Solo is activated, one of the four bands is highlighted indicating the selected band. You select other bands by clicking on them.

The parameters

The following tables list all parameters available in QuadraFuzz.

The parameters in the main window are as follows:

Parameter	Description
Gain dial	This dial can be found in the lower left corner of the QudraFuzz window. You can use it to control the level of the overall input signal before distortion.
Filterbank dials: Low/Low Mid/ High Mid/High	These dials are used to control the output level of the corresponding frequency band <i>after</i> distortion. Values between +/- 12 dB can be set for each band.
Presets fader	This is used to select one of the available presets. To select a new preset, click on the fader handle and drag horizontally.
Output dial	This controls the overall output level.
Over LED	When lit, this indicates that the total input signal level exceeds 0 dB. This LED does not refer to the output level but solely to the input level before distortion. Levels above 0 dB are subject to strict limiting and cause signal clipping. As this is sometimes what you want, QuadraFuzz also offers this option.
Edit button	By clicking on this button, located in the lower right corner of the main window, you can open or close the editor window.

The parameters in the edit window are as follows:

Parameter	Description
Create	If you click on this, a dialog will open where you can add and name a new preset to the preset set currently in memory. The presets are stored with the project – to make a preset available in other projects you use the File pop-up menu as usual.
Delete	This deletes the selected preset from the preset set currently in memory. If you click on the button, a dialog appears where you can confirm or cancel the action.
Solo	This mutes all frequency bands except the selected band.
Shape buttons	The available distortion characteristics (from bottom to top) create effects from a slight distortion up to a trashy hardcore sound.
Frequency band display	Here you control the level and bandwidth for the four bands, see above.

SPL DeEsser (Cubase SX only)



A de-esser is used to reduce excessive sibilance, primarily for vocal recordings. Basically, it is a special type of compressor that is tuned to be sensitive to the frequencies produced by the "s" sound, hence the name de-esser. Close proximity microphone placement and equalizing can lead to situations where the overall sound is just right, but there is a problem with sibilants. Conventional compression and/or equalizing will not easily solve this problem, but a de-esser can.

The SPL DeEsser has the following parameters:

Parameter	Values	Description
S-Reduction	0 - 10	Controls the intensity of the de-essing effect. We recommend that you start with a value between 4 and 7.
Level display		Indicates the dB value by which the level of the sibilant or s-frequency is reduced. The display shows values between 0 dB (no reduction) and minus 20 dB (the s-frequency level is lowered by 20 dB). Each segment in the display represents a level reduction of 2 dB.
Auto Threshold	On/Off	See separate description below.
Male/Female	On/Off	This sets the s-frequency and sibilant recognition to the characteristic frequency ranges of the female or male voice. The center frequency of the bandwidth at which the SPL DeEsser operates is located in the 7 kHz range for the female voice and in the 6 kHz range for the male voice.

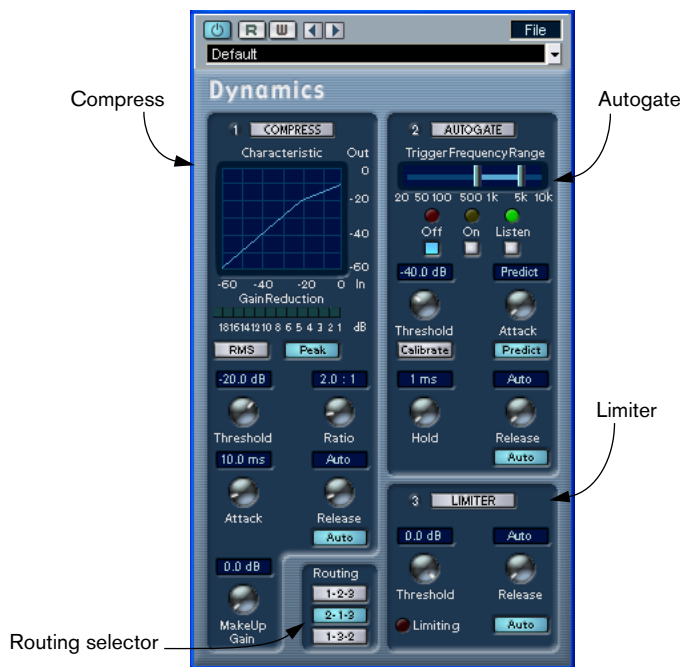
About the Auto Threshold function

Conventional de-essing devices all have a threshold parameter. This is used to set a threshold for the incoming signal level, above which the device starts to process the signal. The SPL DeEsser however has been designed for utmost ease-of-use. With Auto Threshold on (the button is blue) it automatically and constantly readjusts the threshold to achieve an optimum result. If you still wish to determine for yourself at which signal level the SPL DeEsser should start to process the signal, deactivate the Auto Threshold switch. The SPL DeEsser will then use a fixed threshold.

When recording a voice, usually the de-esser's position in the signal chain is located after the microphone pre-amp and before a compressor/limiter. This is useful, as it keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics by reacting to excessive sibilants and s-frequencies.

The Auto Threshold function keeps the processing on a constant level. The input threshold value is automatically and constantly adjusted to the audio input level. Even level differences of say 20 dB do not have a negative impact on the result of the processing. The input levels may vary, but processing remains constant.

Dynamics



Dynamics is an advanced dynamics processor. It combines three separate processors: Autogate, Compress and Limit, covering a variety of dynamic processing functions. The window is divided into three sections, containing controls and meters for each processor.

Activating the individual Processors

You activate the individual processors by clicking on their labels. Activated processors have highlighted labels.

Autogate section

Gating, or noise gating, is a method of dynamic processing that silences audio signals below a certain set threshold level. As soon as the signal level exceeds the set threshold, the gate opens to let the signal through. Autogate offers all the features of a standard noise gate, plus some very useful additional features, such as auto-calibration of the threshold setting, a look-ahead predict function, and frequency selective triggering. Available parameters are as follows:

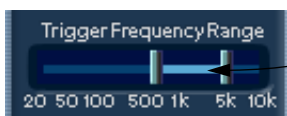
Parameter	Values	Description
Threshold	-60 - 0dB	This setting determines the level where Autogate is activated. Signal levels above the set threshold trigger the gate to open, and signal levels below the set threshold will close the gate.
Attack	0,1 -100 ms or "Predict mode"	This parameter sets the time it takes for the gate to open after being triggered. If the Predict button is activated, it will ensure that the gate will already be open when a signal above the threshold level is played back. Autogate manages this by "looking ahead" in the audio material, checking for signals loud enough to pass the gate.
Hold	0 - 1000 ms	This determines how long the gate stays open after the signal drops below the threshold level.
Release	10 - 1000 ms or "Auto"	This parameter sets the amount of time it takes for the gate to close (after the set hold time). If the "Auto" button is activated, Autogate will find an optimal release setting, depending on the audio program material.

Trigger Frequency Range

Autogate has a feature that allows the gate to be triggered only by signals within a specified frequency range. This is a most useful feature because it lets you filter out parts of the signal that might otherwise trigger the gate in places you don't want it to, thus allowing more control over the gate function. The Trigger Frequency Range function is set using the control in the upper part of the Autogate panel, and the buttons located below it.

The basic operation of the Trigger Frequency Range function is as follows:

1. While playing back audio, click the “Listen” button.
You will now monitor the audio signal, and the gate will be bypassed.
2. While listening, drag the two handles in the Trigger Frequency Range window to set the frequency range you wish to use to trigger the gate.
You will hear the audio being filtered as you move the handles.
 - Dragging the left handle to the right will progressively cut frequencies starting from the low end of the frequency spectrum.
 - Dragging the right handle to the left will progressively cut frequencies starting from the high end of the frequency spectrum.



The frequency range between the two handles will be used to trigger the gate.

3. After setting the frequency range, click the “On” button.
Autogate will now use the selected frequency range as the trigger input.
4. To disable the Trigger Frequency Range function, click the “Off” button.
Autogate will now use the unfiltered audio signal as the trigger input.

Calibrate Function

This function, activated by using the Calibrate button located below the Threshold knob, is used to automatically set the threshold level. It is especially useful for material with consistent inherent background noise, like tape hiss. This may most of the time be masked by the audio content, but becomes noticeable during silent passages. Use it as follows:

1. Find a part of the audio material, preferably not too short, where only the background noise is heard.
If you can only find a short background noise section, try looping it.
2. Play it back, and click on the Calibrate button.
The button will blink for a few seconds, and then automatically set the threshold so that the noise will be silenced (gated) during passages where there is no other signal present.

Compress section

Compress reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. Compress functions like a standard compressor with separate controls for threshold, ratio, attack, release and make-up gain parameters. Compress features a separate display that graphically illustrates the compressor curve shaped according to the Threshold, Ratio and MakeUp Gain parameter settings. Compress also features a Gain Reduction meter that shows the amount of gain reduction in dB, and a program dependent Auto feature for the Release parameter.

The available parameters have the following functionality:

Parameter	Values	Description
Threshold	-60 - 0dB	This setting determines the level where Compress “kicks in”. Signal levels above the set threshold are affected, but signal levels below are not processed.
Ratio	1:1 - 8:1	Ratio determines the amount of gain reduction applied to signals over the set threshold. A ratio of 3:1 means that for every 3 dB the input level increases, the output level will increase by only 1 dB.
Attack	0.1-100 ms	This determines how fast Compress will respond to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) will pass through unprocessed.
Release	10-1000ms or “Auto mode”	Sets the amount of time it takes for the gain to return to its original level when the signal drops below the Threshold level. If the “Auto” button is activated, Compress will automatically find an optimal release setting that varies depending on the audio program material.
MakeUp Gain	0 - 24dB	This parameter is used to compensate for output gain loss, caused by compression.
Compressor Mode	RMS/Peak	RMS mode operates using the average power of the audio signal as a basis, whereas Peak mode operates more on peak levels. As a general guideline, RMS mode works better on material with few transients such as vocals, and Peak mode better for percussive material, with a lot of transient peaks.

Limiter section

Limiter is designed to ensure that the output level never exceeds a certain set output level, to avoid clipping in following devices. Conventional limiters usually require very accurate setting up of the attack and release parameters, to totally avoid the possibility of the output level going beyond the set threshold level. Limiter adjusts and optimizes these parameters automatically, according to the audio material. You can also adjust the Release parameter manually. The available parameters are the following:

Parameter	Values	Description
Threshold	-12 - 0dB	This setting determines the maximum output level. Signal levels above the set threshold are affected, but signal levels below are left unaffected.
Release	10-1000ms or "Auto mode"	This parameter sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the "Auto" button is activated, Limiter will automatically find an optimal release setting that varies depending on the audio program material.

Routing section



In the Routing section you can set the signal flow order for the three processors. Changing the order of the processors can produce different results, and the available options allow you to quickly compare what works best for a given situation. Beside each processor label is a number. These numbers are used to represent the signal flow options shown in the Routing section. There are three routing options:

- 1-2-3 (Compress-Gate-Limit)
- 2-1-3 (Gate-Compress-Limit)
- 1-3-2 (Compress-Limit-Gate)

VST Dynamics



The VST Dynamics plug-in is similar to the Dynamics plug-in (see [page 14](#)), but with the following important differences:

- VST Dynamics has two additional modules: Auto Level and Soft Clip.
- The signal flow is fixed, in the order AutoGate-AutoLevel-Compressor-SoftClip-Limiter.
- VST Dynamics has a higher inherent latency - signals will be delayed when passing through the plug-in.

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- ❑ While Cubase SX/SL automatically compensates for this latency when the plug-in is used as an insert effect for an audio track, this is not true for Group channels, VST Instrument channels or ReWire channels. Therefore, you should only use the VST Dynamics plug-in as an insert effect for audio track (disk) channels (and possibly as a Master effect, if you're only using a single stereo output bus).
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Activating the individual processors

You activate the individual processors by clicking on their labels. Activated processors have highlighted labels. You can activate as many processors as you want, but remember that not all processors are designed to work together. For example, “Limit” and “SoftClip” are both designed to ensure that the output never exceeds 0dB, but achieves this in different ways. To have both of them activated would be unnecessary.

- To turn off all activated VST Dynamics processors, click the lit On button to the right in the panel.
Clicking the button again activates the same configuration of processors.

AutoGate section

This is exactly the same section as the AutoGate in the Dynamics plug-in. See [page 15](#) for details.

AutoLevel section

AutoLevel reduces signal level differences in audio material. It can be used to process recordings where the level unintentionally varies. It will boost low levels and attenuate high level audio signals. Only levels above the set threshold will be processed, so low level noise or rumble will not be boosted. If the input level is greater than 0dB, AutoLevel will react very fast, because it “looks ahead” in the audio material for strong signal levels and can attenuate levels before they occur, thus reducing the risk of signal clipping. AutoLevel has the following parameters:

Parameter	Values	Explanation
Threshold	-90 to -10dB	Only levels stronger than the set threshold will be processed.
Reaction Time Switch	Slow, Mid, Fast	This parameter sets the amount of time it takes for AutoLevel to adjust the gain. Set this according to whether the program level changes suddenly or over a length of time.

Compressor section

This is exactly the same section as the Compress section in the Dynamics plug-in. See [page 17](#) for details.

SoftClip section

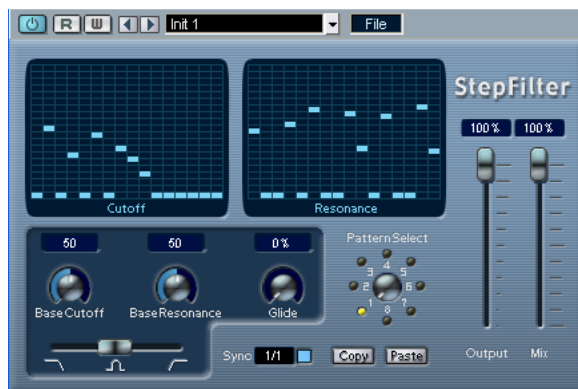
SoftClip is designed to ensure that the output level never exceeds 0dB, like a limiter. SoftClip, however, acts differently compared to a conventional limiter. When the signal level exceeds -6dB, SoftClip starts limiting (or clipping) the signal “softly”, at the same time generating harmonics which add a warm, tubelike characteristic to the audio material. SoftClip is simplicity itself to use as it has no control parameters. The meter indicates the input signal level, and thus the amount of “softclipping”. Levels in the green area (weaker than -6dB) are unaffected, while levels in the yellow-orange-red area indicate the degree of “softclipping”. The deep red meter area to the right indicates input levels higher than 0dB.

- **Avoid feeding SoftClip with excessively high signal levels as audible distortion may occur, although the output level will never exceed 0dB.**

Limiter section

This is exactly the same section as the Limiter in the Dynamics plug-in. See [page 19](#) for details.

StepFilter



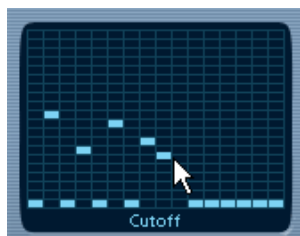
StepFilter is a pattern-controlled multimode filter that can create rhythmic, pulsating filter effects.

General operation

StepFilter can produce two simultaneous 16-step patterns for the filter cutoff and resonance parameters, synchronized to the sequencer tempo.

Setting step values

- Setting step values is done by clicking in the pattern grid windows.
- Individual step entries can be freely dragged up or down the vertical axis, or directly set by clicking in an empty grid box. By click-dragging left or right consecutive step entries will be set to the pointer position.



Setting filter cutoff values in the grid window.

- The horizontal axis shows the pattern steps 1-16 from left to right, and the vertical axis determines the (relative) filter cutoff frequency and resonance setting.
The higher up on the vertical axis a step value is entered, the higher the relative filter cutoff frequency or filter resonance setting.
- By starting playback and editing the patterns for the cutoff and resonance parameters, you can hear how your filter patterns affect the sound source connected to StepFilter directly.

Selecting new patterns

- Created patterns are saved with the song, and up to 8 different cutoff and resonance patterns can be saved internally.
Both the cutoff and resonance patterns are saved together in the 8 Pattern memories.
- To select new patterns you use the pattern selector.
New patterns are all set to the same step value by default.



Pattern Selector

Using pattern copy and paste to create variations

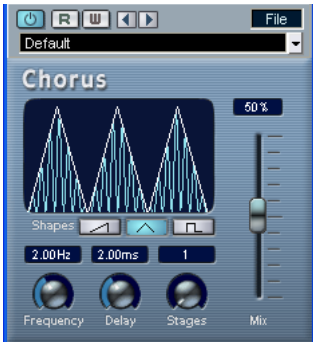
You can use the Copy and Paste buttons below the pattern selector to copy a pattern to another Pattern memory location, which is useful for creating variations on a pattern.

- Click the Copy button with the pattern you wish to copy selected, select another pattern memory location, and click Paste.
The pattern is copied to the new location, and can now be edited to create variations using the original pattern as a starting point.

StepFilter parameters:

Parameter/Value	Description
Base Cutoff	This sets the base filter cutoff frequency. Cutoff values set in the Cutoff grid window are values <i>relative</i> to the Base Cutoff value.
Base Resonance	This sets the base filter resonance. Resonance values set in the Resonance grid window are values <i>relative</i> to the Base Resonance value. Note that very high Base Resonance settings can produce loud ringing effects at certain frequencies.
Glide	This will apply glide between the pattern step values, causing values to change more smoothly.
Filter Mode	This slider selects between lowpass (LP), bandpass (BP) or highpass (HP) filter modes (from left to right respectively).
Sync 1/1-1/32 (Straight, Triplet or Dotted)	This sets the pattern beat resolution, i.e. what note values the pattern will play in relation to the tempo.
Mix	Adjusts the mix between dry and processed signal.
Gain	Sets the overall volume.

Chorus



The Chorus plug-in adds short delays to the signal, and pitch modulates the delayed signals to produce a “doubling” effect. The parameters are as follows:

Parameter	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect. If Chorus is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Shapes	Saw/ Triangle/ Pulse	This sets the modulation waveform. Triangle produces smooth modulation, saw produces ramp shaped modulation and pulse waveform produces stepped modulation.
Frequency	0-5Hz	This sets the modulation rate.
Delay	0-5ms	This controls the depth of the Chorus effect.
Stages	1-3	This adds one or two more delay taps, producing a thicker, multi-layered chorus effect.

- **Note that clicking and dragging in the display allows you to adjust the Frequency and Delay parameters at the same time!**

Flanger



Flanger is a classic flanger effect with stereo enhancement. Parameters are as follows:

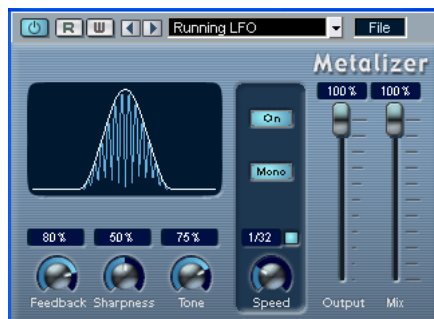
Parameter	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect. If the Flanger is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Tempo sync on/off		The button above the Rate knob is used to switch tempo sync on or off. The button is blue when tempo sync is on, and white when it is off.
Rate	1/1 - 1/32, 1/1 -1/32 Triplet, 1/1 -1/32 Dotted or 0 - 5Hz	If tempo sync is on, this is where you specify the base note value for tempo syncing the flanger sweep. If tempo sync is off, the sweep rate can be set freely with the Rate knob, without sync to tempo.
Tempo Sync knob	x1 to x10	This is the note value multiplier for the flanger sweep when tempo sync is used.
Shape Sync knob	1-16	This changes the shape of the modulating waveform, altering the character of the flanger sweep.
Feedback	0-100%	This determines the character of the flange effect. Higher settings produce a more “metallic” sounding sweep.
Depth	0-100%	This sets the depth of the modulation sweep.
Delay	0-100ms	This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.

Parameter	Values	Description
Stereo Basis	0-100%	This sets the stereo width of the effect. 0% is mono, 50% original stereo, and 100% maximum stereo enhancement.

You can also change parameters in the graphic display window. This works as follows:

- If tempo sync is on, you can set the base note value by clicking the waveform and dragging left and right.
When tempo sync is off, this sets the Rate parameter.
- You can set the Depth parameter by clicking the waveform and dragging up and down.
This means you can freely adjust Rate and Depth at the same time by clicking and dragging.
- By click-dragging the green/blue line in the display left or right you can change the Stereo Basis parameter.

Metalizer



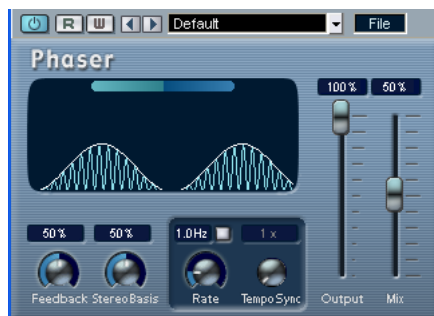
The Metalizer feeds the audio signal through a variable frequency filter, with tempo sync or time modulation and feedback control.

Parameter	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect. If Metalizer is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Tempo sync on/off		The button above the Speed knob is used to switch tempo sync on or off. The button is blue when tempo sync is on, and white when it is off.
Speed	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0 - 10Hz	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect. Note that there is no note value modifier for this effect. If tempo sync is off, the modulation speed can be set freely with the Speed knob, without sync to tempo.
On button	On/Off	Turns filter modulation on and off. When turned off, the Metalizer will work as a static filter.
Mono button	On/Off	Determines whether the output of the Metalizer will be in stereo or mono.
Sharpness	0-100%	Governs the character of the filter effect. The higher the value, the narrower the affected frequency area, producing sharper sound and a more pronounced effect.

Parameter	Values	Description
Tone	0-100%	Governs the feedback frequency. The effect of this will be more noticeable with high Feedback settings.
Feedback	0-100%	Sets the amount of feedback. Higher values produces a more “metallic” sound.

- **Note that clicking and dragging in the display allows you to adjust the Sharpness and Tone parameters at the same time!**

Phaser



The Phaser plug-in produces the classic “swooshing” sound that characterizes phasing. It works by shifting the phase of the signal and adding it back to the original signal, causing partial cancellation of the frequency spectrum.

Parameter	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect. If the Phaser is used as a send effect, this should be set to maximum as you can control the dry/ effect balance with the send.
Tempo sync on/off		The button above the Rate knob is used to switch tempo sync on or off. The button is blue when tempo sync is on, and white when it is off.
Rate	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0 - 5Hz	If tempo sync is on, this is where you specify the base note value for tempo syncing the Phaser sweep. If tempo sync is off, the sweep rate can be set freely with the Rate knob, without sync to tempo.
Feedback	1-100%	This sets the amount of feedback. A higher value produces a more pronounced effect.
TMP Sync knob	x1 to x10	This is the note value multiplier for the Phaser sweep when tempo sync is used.
Stereo Basis	0-100%	This sets the stereo width of the effect. 0% is mono, 50% original stereo, and 100% maximum stereo enhancement.

You can also change parameters in the graphic display window. This works as follows:

- If tempo sync is on, you can set the base note value by clicking the waveform and dragging left and right.
When tempo sync is off, this sets the Rate parameter.
- You can set the Feedback parameter by clicking the waveform and dragging up and down.
This means you can freely adjust the Rate and Feedback at the same time by clicking and dragging.
- By click-dragging the blue/green line in the display left or right you can change the Stereo Basis parameter.

Ringmodulator



The Ringmodulator can produce complex, bell-like enharmonic sounds. Ring modulators work by multiplying two audio signals. The ring modulated output contains added frequencies generated by the sum of, and the difference between, the frequencies of the two signals.

The Ringmodulator has a built-in oscillator that is multiplied with the input signal to produce the effect.

Parameters:

Parameter	Description
Oscillator LFO Amount	LFO Amount controls how much the oscillator frequency is affected by the LFO.
Oscillator Env. Amount	Env. Amount controls how much the oscillator frequency is affected by the Envelope (which is triggered by the input signal). Positive and negative values can be set, with center position representing no modulation. Left of center, a loud input signal will decrease the oscillator pitch, whereas right of center the oscillator pitch will increase when fed a loud input.
Oscillator Wave	Selects the oscillator waveform; square, sine, saw or triangle.
Oscillator Range	Determines the frequency range of the oscillator in Hz.
Oscillator Frequency	Sets the oscillator frequency +/- 2 octaves within the selected range.

Parameter	Description
Roll-Off	Cuts high frequencies in the oscillator waveform, to soften the overall sound. This is best used when harmonically rich waveforms are selected (e.g. square or saw).
LFO Waveform	Selects the LFO waveform; square, sine, saw or triangle.
LFO Speed	Sets the LFO Speed.
LFO Env. Amount	Controls how much the input signal level – via the envelope generator – affects the LFO speed. Positive and negative values can be set, with center position representing no modulation. Left of center, a loud input signal will slow down the LFO, whereas right of center a loud input signal will speed it up.
Invert Stereo	This inverts the LFO waveform for the right channel of the oscillator, which produces a wider stereo perspective for the modulation.
Envelope Generator	The Envelope Generator section controls how the input signal is converted to envelope data, which can then be used to control oscillator pitch and LFO speed. It has two main controls: Attack sets how fast the envelope output level rises in response to a rising input signal. Decay controls how fast the envelope output level falls in response to a falling input signal.
Lock L<R	When this switch is enabled, the L and R input signals are merged, and produce the same Envelope output level for both oscillator channels. When disabled, each channel has its own Envelope, which affect the two channels of the oscillator independently.
Mix	Adjusts the mix between dry and processed signal.
Output	Sets the overall volume.

Rotary



The Rotary plug-in simulates the classic effect of a rotary speaker. A rotary speaker cabinet features variable speed rotating speakers to produce a swirling chorus effect, commonly used with organs. Rotary features all the parameters associated with the real thing. The included presets provide good starting points for further tweaking of the numerous parameters.

The parameters are as follows:

Parameter	Values	Description
Speed	STOP/SLOW/FAST	This controls the speed of the Rotary.
MIDI Ctrl	Controllers [1] to [16]	Selects the MIDI Continuous Controller for the Speed parameter. See page 36 .
Mode	Switched/Variable	Selects whether the SLOW/FAST speed setting is a switch, or a variable control. When switch mode is selected, and Pitch Bend is the controller, the speed will switch with an up or down flick of the bender. Other controllers switch at 64.
Overdrive	0-100%	Applies a soft overdrive or distortion.
Crossover Freq.	200-3000Hz	Sets the crossover frequency between the low and high frequency loudspeakers.
Mic Angle	0-180 degrees	Sets the simulated microphone angle. 0=mono, 180=one mic on each side.
Mic Distance	1-36 Inches	Sets the simulated microphone distance from the speaker.

Parameter	Values	Description
Low Rotor Amp Mod.	0-100%	Adjusts amplitude modulation depth.
Low Rotor Mix Level	0-200%	Adjusts overall bass level.
Hi Rotor Amp Mod.	0-100%	High rotor amplitude modulation.
Hi Rotor Freq. Mod.	0-100%	High rotor frequency modulation.
Phasing	-100 to 100	Adjusts the amount of phasing in the sound of the high rotor.
Hi Slow	0.0-720 rpm	Fine adjustment of the high rotor SLOW speed.
Hi Rate	0.0-720 rpm/sec	Fine adjustment of the high rotor acceleration time.
Hi Fast	0.0-720 rpm	Fine adjustment of the high rotor FAST speed.
Lo Slow	0.0-720 rpm	Fine adjustment of the low rotor SLOW speed.
Lo Rate	0.0-720 rpm/sec	Fine adjustment of the low rotor acceleration time.
Lo Fast	0.0-720 rpm	Fine adjustment of the low rotor FAST speed.
Output		Adjusts the overall output level.
Mix		Adjusts the mix between dry and processed signal.

Directing MIDI to the Rotary

For real-time MIDI control of the Speed parameter, MIDI must be directed to the Rotary.

- Whenever the Rotary has been selected as a Send or Insert effect, it will be available on the “out:” pop-up menu for MIDI Tracks. If Rotary is selected on the “out:” menu, MIDI will be directed to the plug-in from the selected Track.

Symphonic



The Symphonic plug-in combines a stereo enhancer, an auto-panner synchronized to tempo and a chorus-type effect. For best results, apply the Symphonic effect to stereo signals.

The parameters are as follows:

Parameter	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect. If Symphonic is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.
Tempo sync on/off		The button below the Temp sync knob is used to switch tempo sync on or off. The button is blue when tempo sync is on, and white when it is off.
Tempo Sync pop-up	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted	If tempo sync is on, this is where you specify the base note value for tempo syncing the auto-panning.
Tempo Sync knob	x1 to x10	This is the note value multiplier, determining the timing of the auto-panning.
Delay	0-100 ms	This determines the delay time and thus the character of the chorus effect, if activated.
Depth	0-100%	This controls the depth of the chorus effect. If you only want to use Symphonic as an auto-panner or a stereo enhancer, set this to 0%.
Rate	0-100%	This sets the modulation rate for the chorus effect, if activated.
Stereo Basis	0-100%	When the Auto-panner is activated, this sets the stereo width of the panning. When the Auto-panner is deactivated (Tempo sync off), this determines the depth of the stereo enhancer effect. 0% is mono, 50% original stereo, and 100% maximum stereo enhancement.

You can also change parameters in the graphic display window. This works as follows:

- You can set the Rate parameter by clicking the waveform and dragging left and right.
- You can set the Depth parameter by clicking the waveform and dragging up and down.
This means you can freely adjust Rate and Depth at the same time by clicking and dragging.
- By click-dragging the green/blue line in the display left or right you can change the Stereo Basis parameter.

Tranceformer



Tranceformer is a ring modulator effect, in which the incoming audio is ring modulated by an internal, variable frequency oscillator, producing new harmonics. A second oscillator can be used to modulate the frequency of the first oscillator, synchronized to the Song tempo if you wish.

The parameters are as follows:

Parameter	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect.
Tone	1 - 5000Hz	Governs the frequency (pitch) of the modulating oscillator.
Tempo sync on/off		The button above the Speed knob is used to switch tempo sync on or off. The button is blue when tempo sync is on, and white when it is off.
Speed	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0 - 10Hz	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect. Note that there is no note value modifier for this effect. If tempo sync is off, the modulation speed can be set freely with the Speed knob, without sync to tempo.
On button	On/Off	Turns modulation of the Pitch parameter on or off.
Mono button	On/Off	Governs whether the effect output will be stereo or mono.
Depth	0-100%	Governs the depth of the pitch modulation.
Waveform buttons	Sine, Square, Saw, Reverse Saw, Triangle	Sets the pitch modulation waveform.

- **Note that clicking and dragging in the display allows you to adjust the Tone and Depth parameters at the same time!**

Bitcrusher



If you're into lo-fi sound, Bitcrusher is the effect for you. It offers the possibility of decimating and truncating the input audio signal by bit reduction, to get a noisy, distorted sound. You can for example make a 24 bit audio signal sound like an 8 or 4 bit signal, or even render it completely garbled and unrecognizable. Bitcrusher can be used as an insert effect or a send effect. You can of course also use it as a master effect, should you so wish.

Parameter	Values	Description
Mode	I,II,III,IV	Select one of four operating modes for the Bitcrusher. Each mode will produce a different sounding result. Modes I and III are nastier and noisier, while modes II and IV are more subtle.
Depth	0-24	Use this to set the desired bit resolution. A setting of 24 gives the highest audio quality, while a setting of 1 will create mostly noise.
Sample Divider	1-65	This sets the amount by which the audio samples are decimated. At the highest setting (65), nearly all of the information describing the original audio signal will be eliminated, turning the signal into unrecognizable noise.
Mix	N/A	This slider regulates the balance between the output from the Bitcrusher and the original audio signal. Drag the slider upwards for a more dominant effect, and drag it downwards if you want the original signal to be more prominent.
Gain	N/A	Governs the output level from the Bitcrusher. Drag the slider upwards to increase the level.

Chopper



Chopper is a combined tremolo and autopan effect. It can use different waveforms to modulate the level (tremolo) or left-right stereo position (pan), either using tempo sync or manual modulation speed settings. Parameters are as follows:

Parameter	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect. If Chopper is used as a send effect, this should be set to maximum.
Tempo sync on/off		The button above the Speed knob is used to switch tempo sync on (blue button) or off (white button).
Speed	1/1 - 1/32, 1/1 - 1/32 Triplet, 1/1 - 1/32 Dotted or 0 - 50Hz	If tempo sync is on, this is where you specify the base note value for tempo-syncing the effect. Note that there is no note value modifier for this effect. If tempo sync is off, the tremolo/auto-pan speed can be set freely with the Speed knob, without sync to tempo.
Stereo/Mono button	Stereo/Mono	Determines whether the Chopper will work as an auto-panner (button set to "Stereo") or a tremolo effect (button set to "Mono").
Waveform buttons	Sine, Square, Saw, Reverse Saw, Triangle	Sets the modulation waveform.
Depth	0-100%	Sets the depth of the Chopper effect. This parameter can also be set by clicking in the graphic display.

Grungelizer



The Grungelizer adds noise and static to your recordings – kind of like listening to a radio with bad reception, or a worn and scratched vinyl record.

Parameter	Values	Description
Crackle	N/A	This adds crackle to create that old vinyl record sound. The farther to the right you turn the dial, the more crackle is added.
RPM switch	33-45-78	When emulating the sound of a vinyl record, this switch lets you set the RPM (revolutions per minute) speed of the record.
Noise	N/A	This dial regulates the amount of static noise added.
Distort	N/A	Use this dial to add distortion.
EQ	N/A	Turn this dial to the right to cut off the low frequencies, and create a more hollow, lo-fi sound.
AC	N/A	This emulates a constant, low hum of AC current.
Frequency switch	50-60 Hz	This sets the frequency of the AC current, and thus the pitch of the AC hum.
Timeline	Today - 1900	This dial regulates the amount of overall effect. The farther to the right (1900) you turn this dial, the more noticeable the effect.

Vocoder



The Vocoder can apply sound/voice characteristics taken from one signal source, called the “modulator” and apply this to another source, called the “carrier”. A typical application of a vocoder is to use a voice as a modulator and an instrument as a carrier, making the instrument “talk”. A vocoder works by dividing the source signal (modulator) into a number of frequency bands. The audio attributes of these frequency bands can then be used to modulate the carrier.

The Vocoder has a built-in carrier (basically a simple polyphonic synthesizer) but you can also use an external carrier, see [page 46](#).

Setting up – using MIDI

In this mode, the Vocoder is set up slightly differently than other plug-in effects. This is because this setup requires both an audio signal (as the modulator source) and a MIDI input (to play the carrier) to function. To set up for using an external carrier, see [page 46](#).

To set up for use, proceed as follows:

1. Select a source for the modulator.
The modulator source can be audio material from any audio track, or even a live audio input routed to an audio track (provided you have a low latency audio card).
- Good modulator source material are talking or singing voices or percussive sounds, e.g. drum loops.
Static pads or soft ambient material are generally less appropriate for use as modulators, but there are no absolute rules as to what could be used as a modulator source.
2. Select the Vocoder as an insert effect for the audio channel with the modulator signal.
3. Make sure that the Vocoder Mode switch is set to “MIDI”.
4. Select a MIDI track.
This can be an empty MIDI track, or a MIDI track containing data, it doesn't matter. However, if you wish to play the Vocoder in real-time – as opposed to having a recorded part playing it – the track has to have monitoring activated (or be record enabled) for the Vocoder to receive the MIDI output.
5. Select “Vocoder” from the MIDI “out:” pop-up menu for the MIDI track.
The MIDI Output from the track is now routed to the Vocoder. There is an indicator on the Vocoder panel below the Mode switches that blinks when receiving MIDI.

That concludes setting up – you are now ready to start vocoding!

What you do next depends on whether you are using live or recorded audio as the modulator source and whether you are using real-time or recorded MIDI as the carrier input. We will assume for the purposes of this manual that you are using recorded audio as the modulator, and play the carrier in real-time.

6. Make sure the MIDI track is record enabled and start playback.
7. Now play a few notes on your MIDI keyboard.
As you can hear, the audio track material, or rather its formant characteristics, is now applied to the Vocoder's built-in sound source!

Setting up – using an external carrier

There are two modes for using an external carrier:

- “Ext” mode is when the carrier and the modulator can be any two audio sources. The synth section is disabled and grayed out when this mode is selected. MIDI input and the Gap Thru Vocoder parameter are also disabled.
- “Ext+MIDI” mode mixes the audio carrier with the Vocoder’s synth sound. This is described on [page 47](#).

To use an external carrier instead of the built-in synth (“Ext mode”), you set up as follows:

1. Create a Group channel from the Add Track submenu on the Project menu.
2. Open an audio file you wish to use as the carrier source and place it on an empty audio track.
3. Pan the audio channel full right in the Mixer or in the Inspector.
4. Route the output of the audio channel to the group.
5. Open an audio file you wish to use as the modulator source and place it on another empty audio track.
Events on the two audio tracks (carrier and modulator) have to play back simultaneously for the Vocoder to work.
6. Pan the modulator audio channel full left in the Mixer or in the Inspector.
7. Route the output of the modulator audio channel to the group.
8. Select the Vocoder as an insert effect for the group channel.
9. Open the Vocoder panel and activate the “Ext.” Mode button.
10. If you now start playback, the carrier channel will be modulated by the modulator channel!
Note that the synth section on the left half of the Vocoder panel and the “Gap Thru” parameter are now disabled.

Setting up – using an external carrier plus MIDI

Setting up is the same as for using an external carrier, except that a MIDI track with its output routed to the Vocoder should also be present. The MIDI track can either play the Vocoder synth in real time or from prerecorded parts. Make sure that monitoring (or record enable) is activated for the track so that the Vocoder synth will receive MIDI played in real time.

- Set up as described, and activate “MIDI+Ext.” mode on the Vocoder panel.
Any incoming MIDI now triggers the Vocoder synth, and the synths output is mixed with the audio carrier signal.

Vocoder parameters

The Vocoder parameters govern the general sound quality of the vocoded sound.

Parameter	Values	Description
Number of Bands	2-24	This governs how many frequency bands the modulator signal is divided into. Fewer bands will provide a thinner more resonant sound, whereas using more bands will make the sound fuller and more intelligible.
Bandwidth	0-100%	This sets the bandwidth for the frequency bands, which affects the overall timbre. Very narrow bandwidth settings will produce a thin, whistle-like sound.
Min./Max. Freq.	40-8000Hz	These parameters set the minimum and maximum frequency limits for the Vocoder, respectively.
log/lin	0-100%	Log/Lin controls how the frequency bands are spaced between the minimum and maximum frequencies. Log = equal spacing in octaves, Lin = equal spacing in Hz. This affects the basic timbre of the Vocoder.
Env.Speed	10-19699ms/ HOLD	This determines the attack and release times of the Vocoder envelope. Fast settings will cause the modulator signal to trigger the Vocoder instantly, longer settings will gradually increase the attack/release times, providing a more subtle Vocoder effect. If set to “HOLD” the modulator is “frozen”, and doesn’t affect the carrier synth at all.

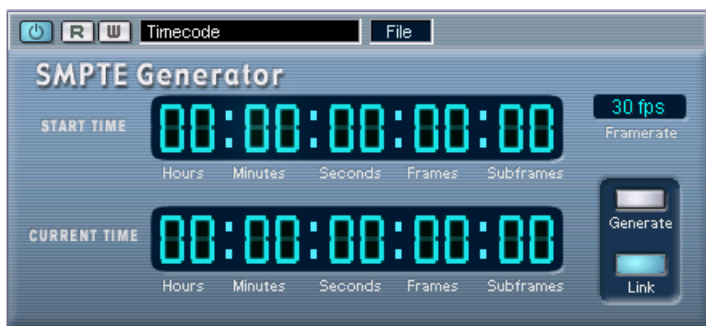
Parameter	Values	Description
High Thru	0-100%	This lets through high frequencies around the “S” frequency from the original input signal while notes are played.
Talk Thru	0-100%	Adjusts the level of the original input signal passed to the Vocoder output while notes are played.
Gap Thru	0-100%	Gap Thru (only available in MIDI mode) sets the level of the original input signal that is passed to the Vocoder output when no MIDI notes are being played. This lets you apply the Vocoder to a vocal track adding vocoded parts just where you want them.
Output	0-100%	This controls the output level of the Vocoder.
Emphasis	0-100%	This is a highpass filter, gradually cutting lower frequencies while letting high frequencies pass.

Vocoder synth parameters

If the built-in synthesizer is the carrier, it is the sound of this instrument that the modulator source is applied to. The synth is polyphonic with up to 8 voices and features 2 oscillators per voice. The synth has the following parameters:

Parameter	Values	Description
Voices	1-8	This sets the number of voices for the synth.
Fine Tune	-100/+100 Cent	Tunes the oscillators in cent (100th of a semitone) steps.
Pitch Bend	1-12 Semitones	Sets the up/down range of the Pitch Bend in semitone steps.
Noise	0-100%	Adds white noise to the sound.
NoiseMod	0-100%	This makes the oscillators modulate the noise level. This gives the noise a rasping sound, turning “sss” into “zzz”.
P.Drift	0-100%	Adds random pitch variation to the oscillators.
P.Glide	0-100%	This makes the pitch glide between notes played. The parameter controls the time it takes for the pitch to glide from one note to the next.
P.Bright	0-100%	This is a lowpass filter that can be used to soften the tone of the oscillators. It does not affect the white noise generator.
P.Detune	12.00 to 0.00	Allows you to detune one of the oscillators in cent steps.
LFO Rate	1-23Hz	Controls the LFO rate (for vibrato).
Vibrato	0-100%	Adds vibrato to the oscillators. This can also be controlled by using the Mod Wheel.

SMPTE Generator (Cubase SX only)



This plug-in is not an effect device. It sends out SMPTE time code to an audio output, allowing you to synchronize other equipment to Cubase SX (provided that the equipment can sync directly to SMPTE time code). This can be very useful if you don't have access to a MIDI to time code converter.

The following items and parameters are available:

- **Generate Button**
Activate this to make the device generate SMPTE time code.
- **Link Button**
This synchronizes the time code output to the Transport time positions. When Generate and Link are both activated, the time code output will exactly match the play position in Cubase SX. Activating the Generate button when Link is off (see below), makes the device send the SMPTE time code in “free run” mode, meaning that it will output continuous time code, independently from the transport status in Cubase SX. If you wish to “stripe” a tape with SMPTE, you should use this mode.
- **Start Time**
This sets the time at which the SMPTE Generator starts, when activated in “free run” mode (Link button off). To change the Start time, click on a digit and move the mouse up or down.
- **Current Time**
When Link is on this shows the current position in Cubase SX. If Link is off it shows the current time of the SMPTE Generator in “free run” mode. This cannot be set manually.

- **Framerate**

This defaults to the frame rate set in the Project Setup. If you wish to generate time code in another frame rate than the Project is currently set to (for example to stripe a tape), you can select another format on the Framerate pop-up (provided that “Link” is off).

Note, however, that for the other device to synchronize correctly with Cubase the frame-rate has to be the same in the Project Setup, the SMPTE Generator and in the receiving device.

Example - Synchronizing a device to Cubase SX

Proceed as follows:

1. **Connect the SMPTE Generator as an insert effect on an audio channel, and route the output of that channel to a separate output.**
Make sure that no other insert or send effects are used on the time code channel. You should also disable EQ, if this is active.
 2. **Connect the corresponding output on the audio hardware to the time code input on the device you wish to synchronize to Cubase.**
Make all necessary settings in the other device, so that it is set to synchronize to incoming timecode.
 3. **Adjust the level of the time code if needed, either in Cubase SX or in the receiving device.**
Activate the SMPTE Generator in “free run” mode to test the level.
 4. **Make sure that the frame rate in the receiving device matches the frame rate set in the SMPTE Generator.**
 5. **Activate the Link button.**
 6. **Activate the Generate Button.**
The SMPTE Generator will now output time code that matches the position of the Cubase SX Transport Panel.
- **Press Play on the Cubase SX Transport Panel.**
The other device is now synchronized and will follow any position changes set with the Cubase SX transport controls.

Reverb A



Reverb A is a reverb plug-in which provides smooth, dense reverb effects. Reverb A has the following parameters:

Parameter	Values	Description
Mix	Dry/Wet	Sets the level balance between the dry signal and the effect (wet). If Reverb A is used as a send effect, this should be set to maximum wet, as you can control the dry/wet balance with the send.
Room Size	20-100	This setting determines the “size” of the simulated room environment.
Predelay	0-100ms	This parameter sets a delay between the direct sound and the reverb effect output. A short predelay before the reverb reduces reverb “clutter” which blurs the sound, and makes the reverb effect more natural-sounding.
Reverb Time	0,2s - forever	This parameter sets the length of the reverb time.
Filter Hi Cut	-15 to 0dB	This filters out high frequencies for the reverb, which can make the reverb sound softer.
Filter Lo Cut	-15 to 0dB	This filters out the lower frequencies for the reverb. It can be used to reduce low frequency “rumble”.

Reverb B



Reverb is used to add ambience and “space” to recordings. The Reverb B effect features the following parameters:

Parameter	Values	Description
Mix	0-100%	Sets the level balance between the dry signal and the effect. If Reverb B is used as a send effect, this should be set to maximum as you can control the dry/ effect balance with the send.
Room Size	0-100%	Governs the “size” of the simulated room environment.
Predelay	0-100%	This parameter sets a delay between the direct sound and the reverb effect output. A short predelay before the reverb reduces reverb “clutter” which blurs the sound, and makes the reverb effect more natural-sounding.
Reverb Time	0-100%	This parameter sets the length of the reverb effect.
Damp	0-100%	This parameter “dampens” the higher frequencies, producing a rounder and smoother sounding reverb.

Mix6To2 (Cubase SX only)



The Mix6To2 Master effect allows you to control the levels of up to six surround channels, and to mix these down to a stereo output. There are ten presets that correspond to the preset (default) surround formats included in the Master Setup. The Mix6To2 allows you to quickly mix down your surround mix format to stereo, and to include parts of the surround channels in the resulting mix.

- Note that the Mix6To2 does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer.

Each of the surround channels has the following parameters:

- Two volume faders that govern the levels of the surround bus to the left and right side of the (master) bus.
- A Link button that links the two volume faders.
- Two Invert buttons allow you to invert the phase of the left and right side of the surround bus.

The Master bus has the following parameters:

- A Link button that links the two Master faders.
- A Normalize button. If activated, the mixed output will be normalized, i.e. the output level will automatically be adjusted so that the loudest signal is as loud as possible without clipping.